INTERNATIONAL APPLICATION PUBLISHED UNDER THE PATENT COOPERATION TREATY (PCT)

(54) Title: A POWER SPECTRAL DENSITY ESTIMATION METHOD AND APPARATUS
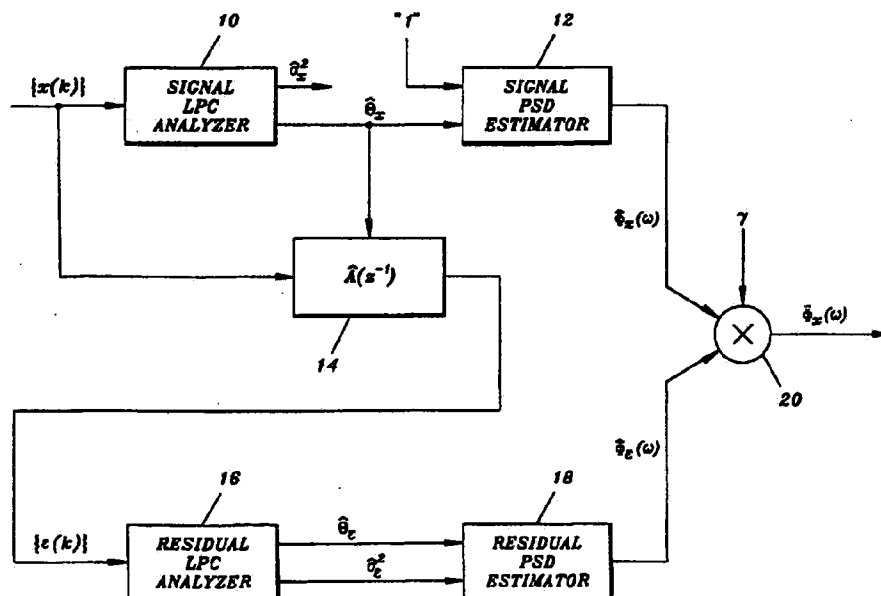
(57) Abstract

A residual error based compensator for the frequency domain bias of an autoregressive spectral estimator is disclosed. LPC analysis (16) is performed on the residual signal and a parametric PSD estimate (18) is formed with the obtained LPC parameters. The PSD estimate of the residual signal multiplies (20) the PSD estimate of the input signal.

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A power spectral density estimation method and
apparatus.

TECHNICAL FIELD

The present invention relates to a bias compensated spectral
estimation method and apparatus based on a parametric auto-
regressive model.

BACKGROUND OF THE INVENTION

The present invention may be applied, for example, to noise
suppression [1, 2] in telephony systems, conventional as well as
cellular, where adaptive algorithms are used in order to model
and enhance noisy speech based on a single microphone measure-
ment.

Speech enhancement by spectral subtraction relies on, explicitly
or implicitly, accurate power spectral density estimates
calculated from the noisy speech. The classical method for
obtaining such estimates is periodogram based on the Fast Fourier
Transform (FFT). However, lately another approach has been
suggested, namely parametric power spectral density estimation,
which gives a less distorted speech output, a better reduction of
the noise level and remaining noise without annoying artifacts
("musical noise"). For details on parametric power spectral
density estimation in general, see [3, 4].

In general, due to model errors, there appears some bias in the
spectral valleys of the parametric power spectral density
estimate. In the output from a spectral subtraction based noise
canceler this bias gives rise to an undesirable "level pumping"
in the background noise.

SUMMARY OF THE INVENTION

An object of the present invention is a method and apparatus that
eliminates or reduces this "level pumping" of the background

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noise with relatively low complexity and without numerical stability problems.

This object is achieved by a method and apparatus in accordance with the enclosed claims.

5   The key idea of this invention is to use a data dependent (or adaptive) dynamic range expansion for the parametric spectrum model in order to improve the audible speech quality in a spectral subtraction based noise canceler.

BRIEF DESCRIPTION OF THE DRAWINGS

10   The invention, together with further objects and advantages thereof, may best be understood by making reference to the following description taken together with the accompanying drawings, in which:

     FIGURE 1   is a block diagram illustrating an embodiment of an
15   apparatus in accordance with the present invention;

     FIGURE 2   is a block diagram of another embodiment of an
                 apparatus in accordance with the present invention;

     FIGURE 3   is a diagram illustrating the true power spectral
                 density, a parametric estimate of the true power
20   spectral density and a bias compensated estimate of
                 the true power spectral density;

     FIGURE 4   is another diagram illustrating the true power
                 spectral density, a parametric estimate of the true
                 power spectral density and a bias compensated
25   estimate of the true power spectral density;

     FIGURE 5   is a flow chart illustrating the method performed
                 by the embodiment of Fig. 1; and

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FIGURE 6   is a flow chart illustrating the method performed
              by the embodiment of Fig. 2.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENTS

Throughout the drawings the same reference designations will be
5   used for corresponding or similar elements.

Furthermore, in order to simplify the description of the present
invention, the mathematical background of the present invention
has been transferred to the enclosed appendix. In the following
description numerals within parentheses will refer to correspon-
10   ding equations in this appendix.

Figure 1 shows a block diagram of an embodiment of the apparatus
in accordance with the present invention. A frame of speech
{x(k)} is forwarded to a LPC analyzer (LPC analysis is described
in, for example, [5]). LPC analyzer 10 determines a set of filter
15   coefficients (LPC parameters) that are forwarded to a PSD
estimator 12 and an inverse filter 14. PSD estimator 12 determi-
nes a parametric power spectral density estimate of the input
frame {x(k)} from the LPC parameters (see (1) in the appendix).
In Fig. 1 the variance of the input signal is not used as an
20   input to PSD estimator 12. Instead a unit signal "1" is forwarded
to PSD estimator 12. The reason for this is simply that this
variance would only scale the PSD estimate, and since this
scaling factor has to be canceled in the final result (se (9) in
the appendix), it is simpler to eliminate it from the PSD
25   calculation. The estimate from PSD estimator 12 will contain the
"level pumping" bias mentioned above.

In order to compensate for the "level pumping" bias the input
frame {x(k)} is also forwarded to inverse filter 14 for forming
a residual signal (see (7) in the appendix), which is forwarded
30   to another LPC analyzer 16. LPC analyzer 16 analyses the residual
signal and forwards corresponding LPC parameters (variance and
filter coefficients) to a residual PSD estimator 18, which forms

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a parametric power spectral density estimate of the residual
signal (see (8) in the appendix).

Finally the two parametric power spectral density estimates of
the input signal and residual signal, respectively, are multi-
5   plied by each other in a multiplier 20 for obtaining a bias
compensated parametric power spectral density estimate of input
signal frame {x(k)} (this corresponds to equation (9) in the
appendix).

Example

10 The following scenario is considered: The frame length N=1024 and
the AR (AR=AutoRegressive) model order p=10. The underlying true
system is modeled by the ARMA (ARMA=AutoRegressive-Moving
Average) process

$$x(k) = \frac{1-z^{-1}+0.9z^{-2}}{1-3.0z^{-1}+4.64z^{-2}-4.44z^{-3}+2.62z^{-4}-0.77z^{-5}} e(k)$$

where e(k) is white noise.

15 Figure 3 shows the true power spectral density of the above
process (solid line), the biased power spectral density estimate
from PSD estimator 12 (dash-dotted line) and the bias compensated
power spectral density estimate in accordance with the present
invention (dashed line). From Fig. 3 it is clear that the bias
20 compensated power spectral density estimate in general is closer
to the underlying true power spectral density. Especially in the
deep valleys (for example for $\omega/(2\pi)\approx0.17$) the bias compensated
estimate is much closer (by 5 dB) to the true power spectral
density.

25 In a preferred embodiment of the present invention a design
parameter γ may be used to multiply the bias compensated
estimate. In Fig. 3 parameter γ was assumed to be equal to 1.
Generally γ is a positive number near 1. In the preferred
embodiment γ has the value indicated in the algorithm section of
30 the appendix. Thus, in this case γ differs from frame to frame.
Fig. 4 is a diagram similar to the diagram in Fig. 3, in which

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the bias compensated estimate has been scaled by this value of γ.

The above described embodiment of Fig. 1 may be characterized as
a frequency domain compensation, since the actual compensation is
performed in the frequency domain by multiplying two power
5   spectral density estimates with each other. However, such an
operation corresponds to convolution in the time domain. Thus,
there is an equivalent time domain implementation of the
invention. Such an embodiment is shown in Fig. 2.

In Fig. 2 the input signal frame is forwarded to LPC analyzer 10
10 as in Fig. 1. However, no power spectral density estimation is
performed with the obtained LPC parameters. Instead the filter
parameters from LPC analysis of the input signal and residual
signal are forwarded to a convolution circuit 22, which forwards
the convoluted parameters to a PSD estimator 12', which forms the
15 bias compensated estimate, which may be multiplied by γ. The
convolution step may be viewed as a polynomial multiplication, in
which a polynomial defined by the filter parameters of the input
signal is multiplied by the polynomial defined by the filter
parameters of the residual signal. The coefficients of the
20 resulting polynomial represent the bias compensated LPC-parame-
ters. The polynomial multiplication will result in a polynomial
of higher order, that is, in more coefficients. However, this is
no problem, since it is customary to "zero pad" the input to a
PSD estimator to obtain a sufficient number of samples of the PSD
25 estimate. The result of the higher degree of the polynomial
obtained by the convolution will only be fewer zeroes.

Flow charts corresponding to the embodiments of Figs. 1 and 2 are
given in Figs. 5 and 6, respectively. Furthermore, the correspon-
ding frequency and time domain algorithms are given in the
30 appendix.

A rough estimation of the numerical complexity may be obtained as
follows. The residual filtering (7) requires $\approx Np$ operations (sum
+ add). The LPC analysis of $e(k)$ requires $\approx Np$ operations to form

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the covariance elements and $\approx p^2$ operations to solve the corresponding set of equations (3). Of the algorithms (frequency and time domain) the time domain algorithm is the most efficient, since it requires $\approx p^2$ operation for performing the convolution. To summarize, the bias compensation can be performed in $\approx 2p(N+p)$ operations/frame. For example, with n=256 and p=10 and 50% frame overlap, the bias compensation algorithm requires approximately $0,5 \times 10^6$ instructions/s.

In this specification the invention has been described with reference to speech signals. However, the same idea is also applicable in other applications that rely on parametric spectral estimation of measured signals. Such applications can be found, for example, in the areas of radar and sonar, economics, optical interferometry, biomedicine, vibration analysis, image processing, radio astronomy, oceanography, etc.

It will be understood by those skilled in the art that various modifications and changes may be made to the present invention without departure from the spirit and scope thereof, which is defined by the appended claims.

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APPENDIX

Consider the real-valued zero mean signal $\{x(k)\}$, $k = 1, \ldots, N$ where N denotes the frame length ($N = 160$, for example). The autoregressive spectral estimator (ARSPE) is given by, see [3, 4]

$$\hat{\Phi}_x(\omega) = \frac{\hat{\sigma}_x^2}{|\hat{A}(e^{i\omega})|^2} \tag{1}$$

where ω is the angular frequency $\omega \in (0, 2\pi)$. In (1), $\hat{A}(z)$ is given by

$$\hat{A}(z) = 1 + \hat{a}_1 z + \cdots + \hat{a}_p z^p \tag{2}$$

where $\hat{\theta}_x = (\hat{a}_1 \cdots \hat{a}_p)^T$ are the estimated AR coefficients (found by LPC analysis, see [5]) and $\hat{\sigma}_x^2$ is the residual error variance. The estimated parameter vector $\hat{\theta}_x$ and $\hat{\sigma}_x^2$ are calculated from $\{x(k)\}$ as follows:

$$\hat{\theta}_x = -\hat{R}^{-1}\hat{r}$$

$$\hat{\sigma}_x^2 = \hat{r}_0 + \hat{r}^T\hat{\theta}_x \tag{3}$$

where

$$\hat{R} = \begin{pmatrix} \hat{r}_0 & \cdots & \hat{r}_{p-1} \\ \vdots & \ddots & \vdots \\ \hat{r}_{p-1} & \cdots & \hat{r}_0 \end{pmatrix} \qquad r = \begin{pmatrix} \hat{r}_1 \\ \vdots \\ \hat{r}_p \end{pmatrix} \tag{4}$$

and, where

$$\hat{r}_k = \frac{1}{N} \sum_{t=1}^{N-k} x(t+k)x(t) \qquad \hat{r}_{-k} = \hat{r}_k \qquad k = 0, \ldots, p \tag{5}$$

The set of linear equations (3) can be solved using the Levinson-Durbin algorithm, see [3]. The spectral estimate (1) is known to be smooth and its statistical properties have been analyzed in [6] for broad-band and noisy narrow-band signals, respectively.

In general, due to model errors there appears some bias in the spectral valleys. Roughly, this bias can be described as

$$\hat{\Phi}_x(\omega) - \Phi_x(\omega) \begin{cases} \approx 0 & \text{for } \omega \text{ such that } \Phi_x(\omega) \approx \max_\omega \Phi_x(\omega) \\ \gg 0 & \text{for } \omega \text{ such that } \Phi_x(\omega) \ll \max_\omega \Phi_x(\omega) \end{cases} \tag{6}$$

where $\hat{\Phi}_x(\omega)$ is the estimate (1) and $\Phi_x(\omega)$ is the true (and unknown) power spectral density of $x(k)$.

In order to reduce the bias appearing in the spectral valleys, the residual is calculated according to

$$\varepsilon(k) = \hat{A}(z^{-1})x(k) \qquad k = 1, \ldots, N \tag{7}$$

Performing another LPC analysis on $\{\varepsilon(k)\}$, the residual power spectral density can be calculated from, cf. (1)

$$\hat{\Phi}_{\varepsilon}(\omega) = \frac{\hat{\sigma}_{\varepsilon}^2}{|\hat{B}(e^{i\omega})|^2} \tag{8}$$

where, similarly to (2), $\hat{\theta}_{\varepsilon} = (\hat{b}_1 \cdots \hat{b}_q)^T$ denotes the estimated AR coefficients and $\hat{\sigma}_{\varepsilon}^2$ the error variance. In general,the model order $q \neq p$, but here it seems reasonable to let $p = q$. Preferably $p \approx \sqrt{N}$, for example N may be chosen around 10.

In the proposed frequency domain algorithm below, the estimate (1) is compensated according to

$$\bar{\Phi}_x(\omega) = \frac{\gamma}{\hat{\sigma}_x^2} \cdot \hat{\Phi}_{\varepsilon}(\omega) \cdot \hat{\Phi}_x(\omega) \tag{9}$$

where γ (≈ 1) is a design variable. The frequency domain algorithm is summarized in the algorithms section below and in the block diagrams in Fig. 1 and 5.

A corresponding time domain algorithm is also summarized in the algorithms section and in Fig. 2 and 6. In this case the compensation is performed in a convolution step, in which the LPC filter coefficients $\hat{\theta}_x$ are compensated. This embodiment is more efficient, since one PSD estimation is replaced by a less complex convolution. In this embodiment the scaling factor γ may simply be set to a constant near or equal to 1. However, it is also possible to calculate γ for each frame, as in the frequency domain algorithm by calculating the root of the characteristic polynomial defined by $\hat{\theta}_{\varepsilon}$ that lies closest to the unit circle. If the angle of this root is denoted $\bar{\omega}$, then

$$\max_k \hat{\Phi}_{\varepsilon}(k) = \frac{\hat{\sigma}_{\varepsilon}^2}{|\hat{B}(e^{i\bar{\omega}})|^2}$$

ALGORITHMS

INPUTS

x input data $x = (x(1) \cdots x(N))^T$

p LPC model order

OUTPUTS

$\hat{\theta}_x$ signal LPC parameters $\hat{\theta}_x = (\hat{a}_1 \cdots \hat{a}_p)^T$

$\hat{\sigma}_x^2$ signal LPC residual variance

$\hat{\Phi}_x$ signal LPC spectrum $\hat{\Phi}_x = (\hat{\Phi}_x(1) \cdots \hat{\Phi}_x(N/2))^T$

$\bar{\Phi}_x$ compensated LPC spectrum $\bar{\Phi}_x = (\bar{\Phi}_x(1) \cdots \bar{\Phi}_x(N/2))^T$

ε residual $\varepsilon = (\varepsilon(1) \cdots \varepsilon(N))^T$

$\hat{\theta}_\varepsilon$ residual LPC parameters $\hat{\theta}_\varepsilon = (\hat{b}_1 \cdots \hat{b}_p)^T$

$\hat{\sigma}_\varepsilon^2$ residual LPC error variance

γ design variable $(=1/(\max_k \hat{\Phi}_\varepsilon(k))$ in preferred embodiment)

FREQUENCY DOMAIN ALGORITHM

FOR EACH FRAME DO THE FOLLOWING STEPS:

(power spectral density estimation)

$[\hat{\theta}_x, \hat{\sigma}_x^2] := $ LPCanalyze(x, p) signal LPC analysis

$\hat{\Phi}_x := $ SPEC$(\hat{\theta}_x, 1, N)$ signal spectral estimation, $\hat{\sigma}_x^2$ set to 1

(bias compensation)

$\varepsilon := $ FILTER$(\hat{\theta}_x, x)$ residual filtering

$[\hat{\theta}_\varepsilon, \hat{\sigma}_\varepsilon^2] := $ LPCanalyze(ε, p) residual LPC analysis

$\hat{\Phi}_\varepsilon := $ SPEC$(\hat{\theta}_\varepsilon, \hat{\sigma}_\varepsilon^2, N)$ residual spectral estimation

FOR k=1 TO $N/2$ DO   spectral compensation

$\qquad \bar{\Phi}_x(k) := \gamma \cdot \hat{\Phi}_x(k) \cdot \hat{\Phi}_\varepsilon(k)$ $1/(\max_k \hat{\Phi}_\varepsilon(k)) \leq \gamma \leq 1$

END FOR

TIME DOMAIN ALGORITHM

FOR EACH FRAME DO THE FOLLOWING STEPS:

$[\hat{\theta}_x, \hat{\sigma}_x^2] := $ LPCanalyze(x, p) signal LPC analysis

$\varepsilon := $ FILTER$(\hat{\theta}_x, x)$ residual filtering

$[\hat{\theta}_\varepsilon, \hat{\sigma}_\varepsilon^2] := $ LPCanalyze(ε, p) residual LPC analysis

$\hat{\theta} := $ CONV$(\hat{\theta}_x, \hat{\theta}_\varepsilon)$ LPC compensation

$\hat{\Phi} := $ SPEC$(\hat{\theta}, \hat{\sigma}_\varepsilon^2, N)$ spectral estimation

FOR k=1 TO $N/2$ DO

$\qquad \bar{\Phi}_x(k) := \gamma \cdot \hat{\Phi}(k)$ scaling

END FOR

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CLAIMS

1. A power spectral density estimation method, comprising the
steps of:
    performing a LPC analysis on an input signal vector for
5   determining a first set of LPC filter parameters;
    determining a first power spectral density estimate of said
input signal vector based on said first set of LPC filter parame-
ters;
    filtering said input signal vector through an inverse LPC
10 filter determined by said first set of LPC filter parameters for
obtaining a residual signal vector;
    performing a LPC analysis on said residual signal vector for
determining a second set of LPC filter parameters;
    determining a second power spectral density estimate of said
15 residual signal vector based on said second set of LPC filter
parameters; and
    forming a bias compensated power spectral estimate of said
input signal vector that is proportional to the product of said
first and second power spectral estimates.

20 2. The method of claim 1, wherein said product is multiplied by
a positive scaling factor that is less than or equal to 1.

3. The method of claim 2, wherein said scaling factor is the
inverted value of the maximum value of said second power spectral
density estimate.

25 4. The method of claim 1, 2 or 3, wherein said input signal
vector comprises speech samples.

5. A power spectral density estimation method, comprising the
steps of:
    performing a LPC analysis on an input signal vector for
30 determining a first set of LPC filter parameters;
    filtering said input signal vector through an inverse LPC
filter determined by said first set of LPC filter parameters for

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obtaining a residual signal vector;

performing a LPC analysis on said residual signal vector for determining a second set of LPC filter parameters;

convolving said first set of LPC filter parameters with said second set of LPC filter parameters for forming a compensated set of LPC filter parameters;

determining a bias compensated power spectral density estimate of said input signal vector based on said compensated set of LPC filter parameters.

6. The method of claim 5, wherein said bias compensated power spectral density estimate is multiplied by a positive scaling factor that is less than or equal to 1.

7. The method of claim 6, wherein said scaling factor is the inverted value of the maximum value of a power spectral density estimate of said residual signal vector.

8. The method of claim 5, 6 or 7, wherein said input signal vector comprises speech samples.

9. A power spectral density estimation apparatus, comprising:

means (10) for performing a LPC analysis on an input signal vector for determining a first set of LPC parameters;

means (12) for determining a first power spectral density estimate of said input signal vector based on said first set of LPC parameters;

means (14) for filtering said input signal vector through an inverse LPC filter determined by said first set of LPC parameters for obtaining a residual signal vector;

means (16) for performing a LPC analysis on said residual signal vector for determining a second set of LPC parameters;

means (18) for determining a second power spectral density estimate of said residual signal vector based on said second set of LPC parameters; and

means (20) for forming a bias compensated power spectral estimate of said input signal vector that is proportional to the

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product of said first and second power spectral estimates.

10. A power spectral density estimation apparatus, comprising:
     means (10) for performing a LPC analysis on an input signal
vector for determining a first set of LPC filter parameters;
     means (14) for filtering said input signal vector through an
inverse LPC filter determined by said first set of LPC filter
parameters for obtaining a residual signal vector;
     means (16) for performing a LPC analysis on said residual
signal vector for determining a second set of LPC filter
parameters;
     means (22) for convolving said first set of LPC filter
parameters with said second set of LPC filter parameters for
forming a compensated set of LPC filter parameters;
     means (12') for determining a bias compensated power spectral
density estimate of said input signal vector based on said
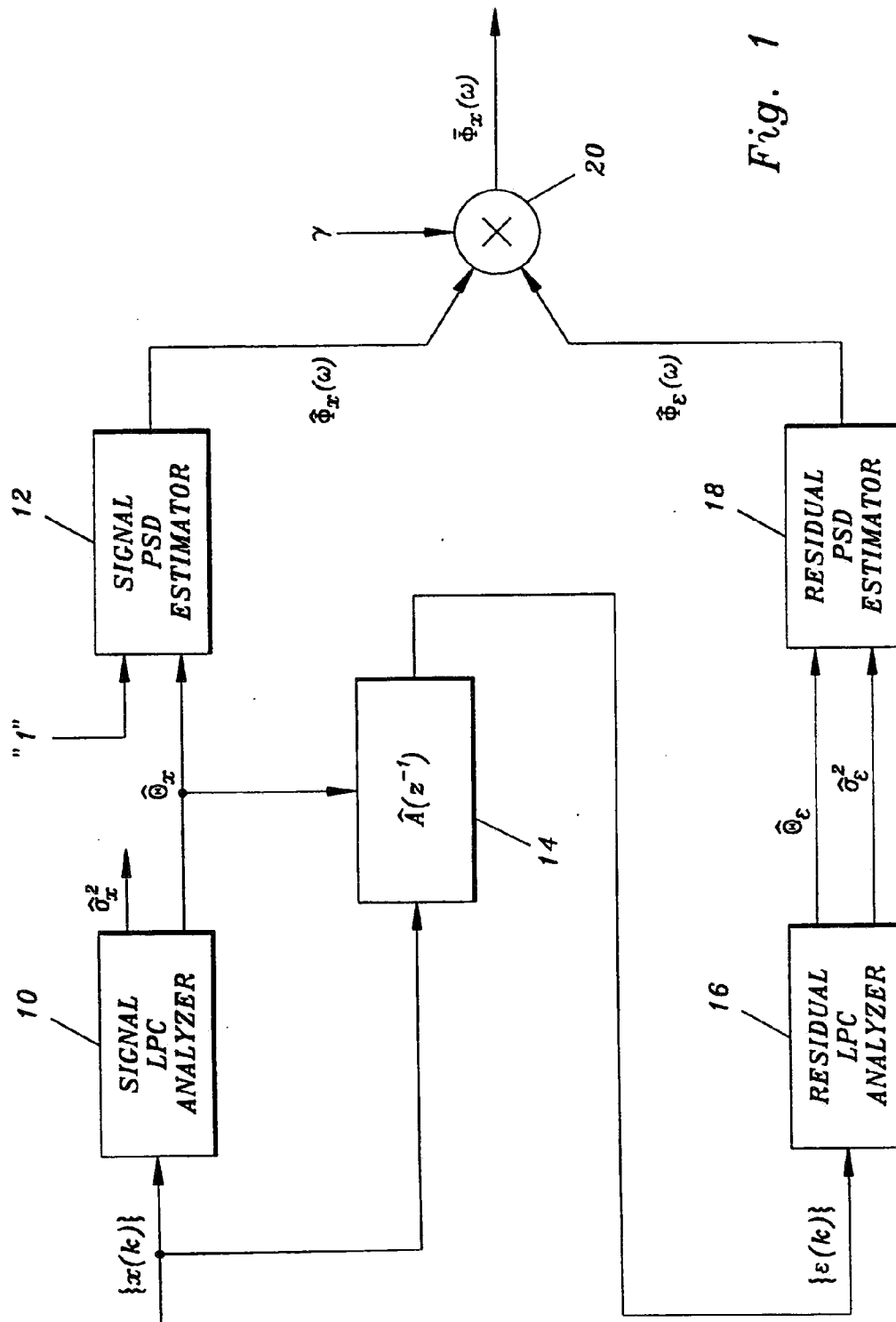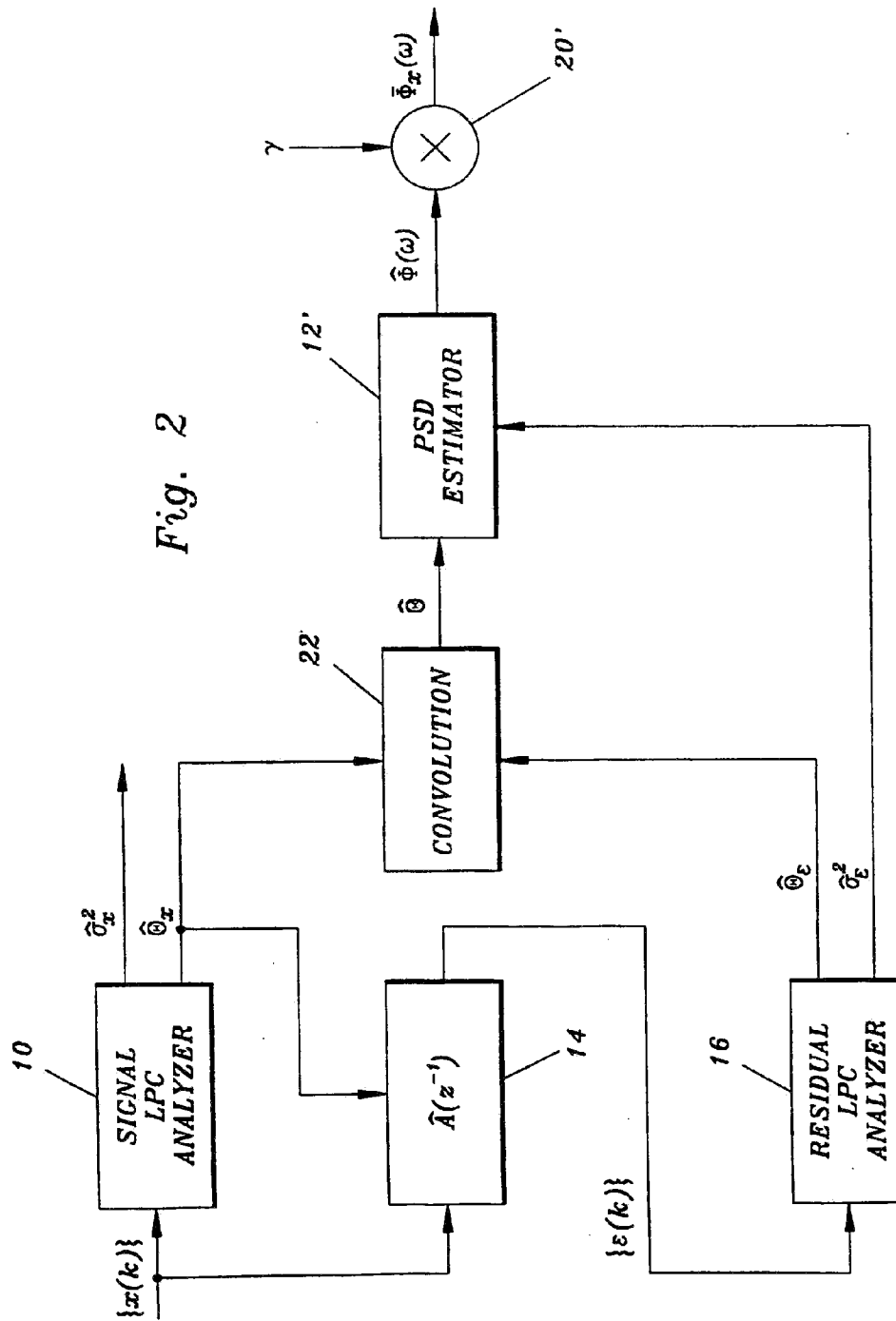compensated set of LPC filter parameters.

Fig. 1

Fig. 2

Fig. 3

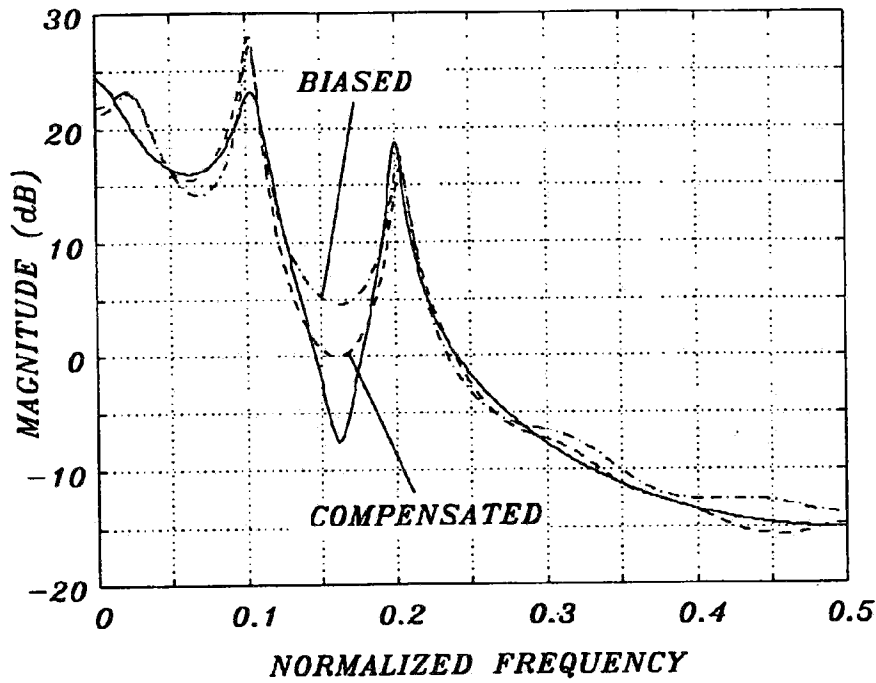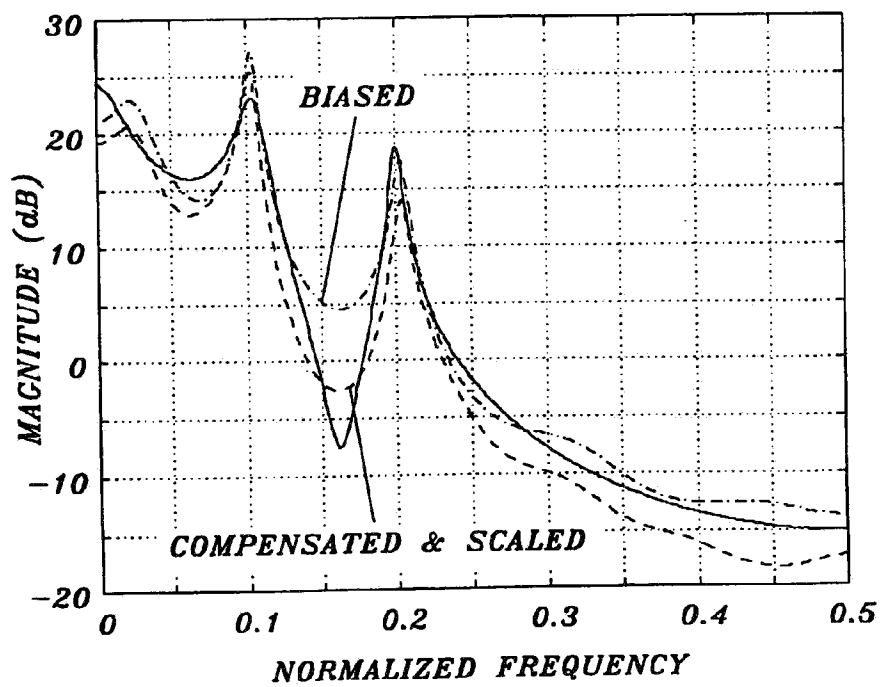
Fig. 4

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Fig. 5

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┌─────────────────────┐
│    LPC ANALYZE      │
│    INPUT SIGNAL     │
└─────────────────────┘
           │
┌─────────────────────┐
│    ESTIMATE PSD     │
│   OF INPUT SIGNAL   │
└─────────────────────┘
           │
┌─────────────────────┐
│      FILTER         │
│    INPUT SIGNAL     │
└─────────────────────┘
           │
┌─────────────────────┐
│    LPC ANALYZE      │
│   RESIDUAL SIGNAL   │
└─────────────────────┘
           │
┌─────────────────────┐
│   ESTIMATE PSD OF   │
│   RESIDUAL SIGNAL   │
└─────────────────────┘
           │
┌─────────────────────┐
│    COMPENSATE       │
│    INPUT SIGNAL     │
│   PSD ESTIMATE      │
└─────────────────────┘
           │
┌ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ┐
│      SCALE          │
│   PSD ESTIMATE      │
└ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ┘
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Fig. 5

Fig. 6

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┌─────────────────────┐
│    LPC ANALYZE      │
│    INPUT SIGNAL     │
└─────────────────────┘
           │
┌─────────────────────┐
│      FILTER         │
│    INPUT SIGNAL     │
└─────────────────────┘
           │
┌─────────────────────┐
│    LPC ANALYZE      │
│   RESIDUAL SIGNAL   │
└─────────────────────┘
           │
┌─────────────────────┐
│    COMPENSATE       │
│    INPUT SIGNAL     │
│   LPC PARAMETERS    │
└─────────────────────┘
           │
┌─────────────────────┐
│    ESTIMATE PSD     │
│  FROM COMPENSATED   │
│   LPC PARAMETERS    │
└─────────────────────┘
           │
┌ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ┐
│      SCALE          │
│   PSD ESTIMATE      │
└ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ┘
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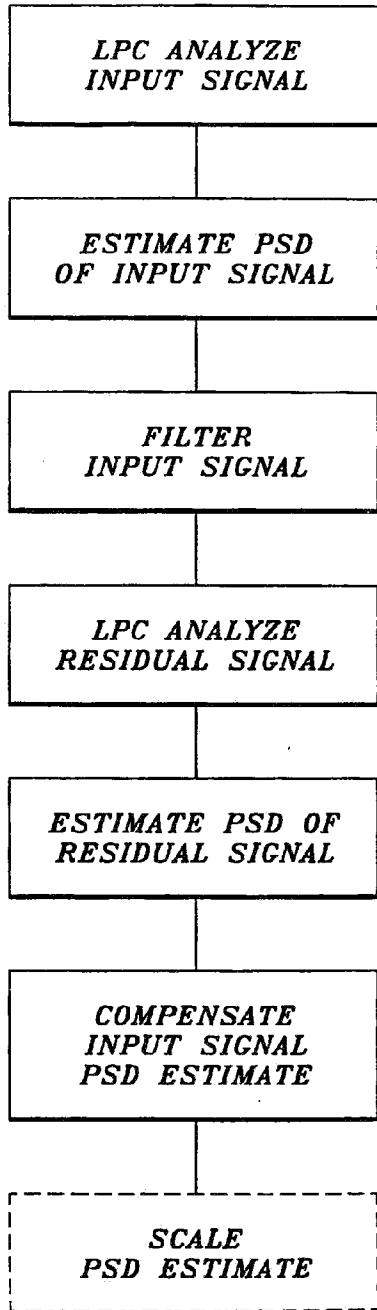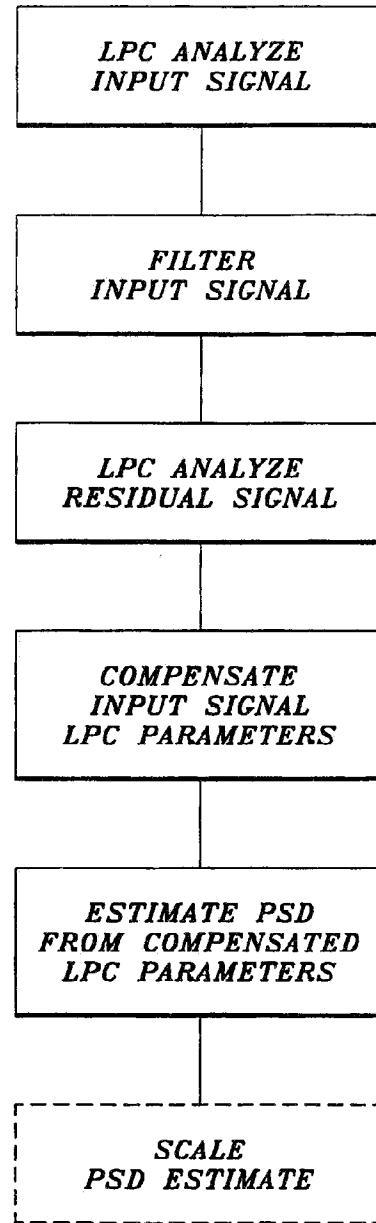
Fig. 6

INTERNATIONAL SEARCH REPORT

International application No.
PCT/SE 96/00753

A. CLASSIFICATION OF SUBJECT MATTER

IPC6: G01R 23/16, H03H 21/00
According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

IPC6: G01R, H03H

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

SE,DK,FI,NO classes as above

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)

DIALOG: WPI, CLAIMS

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	EP 0588526 A1 (NOKIA MOBILE PHONES LTD.), 23 March 1994 (23.03.94), column 3, line 20 - line 39, figure 1, abstract	1,5,9,10
A	US 5208862 A (KAZUNORI OZAWA), 4 May 1993 (04.05.93), abstract	1,5,9,10

☐ Further documents are listed in the continuation of Box C. ☒ See patent family annex.

Date of the actual completion of the international search	Date of mailing of the international search report
9 October 1996	**1 1 -10- 1996**

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Form PCT/ISA/210 (second sheet) (July 1992)

International application No.
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